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In re application of: Jarmo Kuusinen et al
Serial No.: 10/613,986 Group No.:
Filed: July 3, 2003 Examiner:
For: MANAGING A PACKET SWITCHED CONFERENCE CALL

Commissioner for Patents
Alexandria, VA 22313-1450

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Attached please find the certified copy of the foreign application from which priority is claimed for this case:

Country:	WO
Application Number:	PCT/IB02/02625
Filing Date:	July 4, 2002

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CERTIFICATE OF MAILING (37 CFR 1.8a)

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PATENT COOPERATION TREATY (PCT)
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1/4

PCT REQUEST

020599WO

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0	For receiving Office use only	
0-1	International Application No.	PCT/IB 02 / 02625
0-2	International Filing Date	04 JULY 2002 (04.07.02)
0-3	Name of receiving Office and "PCT International Application"	INTERNATIONAL BUREAU OF WIPO PCT International Application
0-4	Form - PCT/RO/101 PCT Request	
0-4-1	Prepared using	PCT-EASY Version 2.92 (updated 01.01.2002)
0-5	Petition The undersigned requests that the present international application be processed according to the Patent Cooperation Treaty	
0-6	Receiving Office (specified by the applicant)	International Bureau of the World Intellectual Property Organization (RO/IB)
0-7	Applicant's or agent's file reference	020599WO
I	Title of Invention	MANAGING A PACKET SWITCHED CONFERENCE CALL
II	Applicant	
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V	Designation of States	
V-1	Regional Patent (other kinds of protection or treatment, if any, are specified between parentheses after the designation(s) concerned)	AP: GH GM KE LS MW MZ SD SL SZ TZ UG ZM ZW and any other State which is a Contracting State of the Harare Protocol and of the PCT EA: AM AZ BY KG KZ MD RU TJ TM and any other State which is a Contracting State of the Eurasian Patent Convention and of the PCT EP: AT BE CH&LI CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR and any other State which is a Contracting State of the European Patent Convention and of the PCT OA: BF BJ CF CG CI CM GA GN GQ GW ML MR NE SN TD TG and any other State which is a member State of OAPI and a Contracting State of the PCT
V-2	National Patent (other kinds of protection or treatment, if any, are specified between parentheses after the designation(s) concerned)	AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH&LI CN CO CR CU CZ DE DK DM DZ EC EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ OM PH PL PT RO RU SD SE SG SI SK SL TJ TM TN TR TT TZ UA UG US UZ VN YU ZA ZM ZW

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V-5	Precautionary Designati n Statement In addition to the designations made under items V-1, V-2 and V-3, the applicant also makes under Rule 4.9(b) all designations which would be permitted under the PCT except any designation(s) of the State(s) indicated under item V-6 below. The applicant declares that those additional designations are subject to confirmation and that any designation which is not confirmed before the expiration of 15 months from the priority date is to be regarded as withdrawn by the applicant at the expiration of that time limit.		
V-6	Exclusion(s) from precautionary designations	NONE	
VI	Priority claim	NONE	
VII-1	International Searching Authority Chosen	European Patent Office (EPO) (ISA/EP)	
VIII	Declarations	Number of declarations	
VIII-1	Declaration as to the identity of the inventor	-	
VIII-2	Declaration as to the applicant's entitlement, as at the international filing date, to apply for and be granted a patent	-	
VIII-3	Declaration as to the applicant's entitlement, as at the international filing date, to claim the priority of the earlier application	-	
VIII-4	Declaration of inventorship (only for the purposes of the designation of the United States of America)	-	
VIII-5	Declaration as to non-prejudicial disclosures or exceptions to lack of novelty	-	
IX	Check list	number of sheets	electronic file(s) attached
IX-1	Request (including declaration sheets)	4	-
IX-2	Description	17	-
IX-3	Claims	4	-
IX-4	Abstract	1	EZABST00.TXT
IX-5	Drawings	3	-
IX-7	TOTAL	29	
	Accompanying items	paper document(s) attached	electronic file(s) attached
IX-8	Fee calculation sheet	✓	-
IX-17	PCT-EASY diskette	-	Diskette
IX-19	Figure of the drawings which should accompany the abstract	3	
IX-20	Language of filing of the international application	English	

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X-1	Signature of applicant, agent or comm n representative	A. Weyres
X-1-1	Name	COHAUSZ & FLORACK
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X-1-3	Capacity	Patent Attorney

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10-1	Date of actual receipt of the purported international application	04 JULY 2002 (04. 07. 02)
10-2	Drawings:	
10-2-1	Received	
10-2-2	Not received	
10-3	Corrected date of actual receipt due to later but timely received papers or drawings completing the purported international application	
10-4	Date of timely receipt of the required corrections under PCT Article 11(2)	
10-5	International Searching Authority	ISA/EP
10-6	Transmittal of search copy delayed until search fee is paid	

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PCT (ANNEX - FEE CALCULATION SHEET)

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(This sheet is not part of and does not count as a sheet of the international application)

0	For receiving Office use only			
0-1	International Application No.			
0-2	Date stamp of the receiving Office			

0-4	Form - PCT/RO/101 (Annex) PCT Fee Calculation Sheet			
0-4-1	Prepared using	PCT-EASY Version 2.92 (updated 01.01.2002)		
0-9	Applicant's or agent's file reference	020599WO		
2	Applicant	NOKIA CORPORATION, et al.		
12	Calculation of prescribed fees	fee amount/multiplier	Total amounts (CHF)	
12-1	Transmittal fee T	⇨	100	
12-2-1	Search fee S	⇨	1.383	
12-2-2	International search to be carried out by	EP		
12-3	International fee Basic fee (first 30 sheets) b1	650		
12-4	Remaining sheets	0		
12-5	Additional amount (X)	15		
12-6	Total additional amount b2	0		
12-7	b1 + b2 = B	650		
12-8	Designation fees Number of designations contained in international application	93		
12-9	Number of designation fees payable (maximum 5)	5		
12-10	Amount of designation fee (X)	140		
12-11	Total designation fees D	700		
12-12	PCT-EASY fee reduction R	-200		
12-13	Total International fee (B+D-R) I	⇨	1.150	
12-17	TOTAL FEES PAYABLE (T+S+I+P)	⇨	2.633	
12-19	Mode of payment	authorization to charge deposit account		
12-20	Deposit account instructions The receiving Office:	International Bureau of the World Intellectual Property Organization (RO/IB)		
12-20-1	Authorization to charge the total fees indicated above.	✓		
12-20-2	Authorization to charge any deficiency or credit any overpayment in the total fees indicated above.	✓		
12-21	Deposit account No.	11486		
12-22	Date	04 July 2002 (04.07.2002)		

PCT (ANNEX - FEE CALCULATION SHEET)

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12-23	Name and signature	COHAUSZ & FLORACK, Alexandra Weyres A. Weyres

VALIDATION LOG AND REMARKS

13-2-3	Validation messages Names	Green? Applicant 1.:Telephone No. missing
		Green? Applicant 1.:Facsimile No. missing
13-2-4	Validation messages Priority	Green? No priority of an earlier application has been claimed. Please verify
13-2-7	Validation messages Contents	Yellow! The power of attorney or a copy of the general power of attorney will need to be furnished unless all applicants sign the request form.
13-2-8	Validation messages Fees	Green? Please confirm that fee schedule utilized is the latest available
13-2-9	Validation messages Payment	Green? Please ensure that you have a valid deposit account with the receiving Office selected.

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July 04, 2002

Managing a packet switched conference call

FIELD OF THE INVENTION

The invention relates to a method for managing a packet switched centralized conference call between a plurality of terminals. The invention relates equally to a conference call server comprising means for managing a centralized conference call and to a terminal comprising means for participating in a centralized conference call.

BACKGROUND OF THE INVENTION

In a conference call, a group of terminal users is connected together in a way that when one of the participating users talks, all other participating users are able to hear the voice of the talking participant. In such a kind of communication, normally only one of the participating users is talking at a time, while the other users are listening. In a centralized conference call, the terminals of the participating users are not connected directly with each other, but via a conference call server. A centralized conference call can be realized for instance by a Voice over Internet Protocol (VoIP) conference call application in the internet or as voice conferencing in Universal Mobile Telecommunication Services (UMTS) network's packet switched domain.

In a VoIP session, the voice data is typically carried by using the Real-time Transport Protocol (RTP) on top of

the Internet Protocol (IP) and the User Datagram Protocol (UDP). RTP has been described in detail in RFC 1889: "RTP: A Transport Protocol for Real-Time Applications", January 1996, by H. Schulzrinne et al.

An end-to-end VoIP connection is often called VoIP tunnel. In a typical centralized conference call set-up, VoIP tunnels are formed between each participating terminal and the conference call server.

For illustration, the tunneling of coded voice in a centralized, RTP based conference call is presented in figure 1.

Figure 1 schematically shows a centralized conference call system in a packet switched domain of UMTS network 11, with a conference call server 12 connected to this network 11 and with a plurality of mobile terminals 13. The mobile terminals 13 are connected to the conference call server via the UMTS network 11 using RTP tunnels 14.

At the terminals 13, voice data produced by the respective user of the terminals 13 is first encoded and then inserted to the payload of RTP packets. There is a multitude of alternative audio coders that can be used to perform the actual voice coding. For example, the Adaptive Multirate (AMR) speech codec, which is specified as the mandatory speech codec for the 3rd generation systems, could be used to compress the speech data carried inside the RTP payload. The coders encode the speech samples to frames, which are then carried over the RTP/UDP/IP protocols via the UMTS network 11 to the conference call server 12.

The conference call server 12 comprises an RTP mixer 15, which receives the incoming RTP packet flows from the connected terminals 13, removes the RTP packaging, combines the flows into a single flow of RTP packets and then sends this flow to each of the terminals 13.

To each RTP packet transmitted between the terminals 13 and the conference call server 12, a header is associated. The structure of this header, which is specified in the above cited RFC 1889, is illustrated in figure 2. The header comprises a field V which identifies the version of the employed RTP and a field P for a padding bit. If the padding bit is set, the packet contains one or more additional padding octets at the end which are not part of the payload. The header further comprises a field X for an extension bit. If the extension bit is set, the fixed header is followed by exactly one header extension. The header moreover comprises a field CC for a Contributing Source (CSRC) count, which contains the number of CSRC identifiers that follow the fixed header, and a field M for a marker bit, the interpretation of the marker being defined by a profile. In addition, the header comprises a field PT for identifying the format of the payload and a field for a Sequence Number, which increments by one for each RTP data packet sent. The Sequence Number may be used by the receiver to detect a packet loss and to restore the packet sequence. The header also comprises a field for a Timestamp, which reflects the sampling instant of the first octet in the RTP data packet.

Furthermore, the RTP packet headers carry a Synchronisation Source (SSRC) identifier and, as

mentioned above with reference to the CC field, a list of Contributing Source (CSRC) identifiers.

The SSRC identifier is used to identify the synchronization source that has transmitted the RTP packet in question. An SSRC identifier which is unique for the respective RTP session is associated randomly to each possible source, i.e. to each of the terminals 13 and to the conference call server 12. Each terminal 13 adds the SSRC identifier associated to it to the SSRC identifier field in the RTP header of each RTP packet it assembles. Equally, the RTP mixer 15 of the conference call server 12 adds the SSRC identifier associated to the conference call server 12 to the SSRC identifier field in the RTP header of each RTP packet leaving the server 12.

The CSRC list is used to identify different sources contributing to an RTP packet and is thus only of relevance for the RTP packets assembled in the conference call server 13. The RTP mixer 15 adds the SSRC identifiers of those terminals 13 contributing to the combined outbound VoIP flow to the CSRC fields of outgoing RTP packets.

In order to enable a control of the VoIP connections using RTP, in addition a Real Time Control Protocol (RTCP) is defined in the above cited RFC 1889. RTCP is used for instance to keep the both ends of a connection informed about the quality of service they are providing and receiving. This information is sent in RTCP sender report (SR) and receiver report (RR) packet types. In addition, the RTP specification defines an RTCP source description (SDES) packet type. RTCP SDES packets can be used by the source to provide more information about

itself. SDES CNAME or NAME packets can be used for example to provide a mapping between the random SSRC identifier and the source identity. SDES CNAME packets are intended for providing canonical end-point identifiers, while SDES NAME packets are intended for providing a real name used to describe the respective source. The RTP mixer 15 is expected to combine SR and RR type RTCP packets from all terminals 13 before forwarding them. The SDES type RTCP packets, in contrast, are forwarded by the RTP mixer 15 to all conference participants 13 without modifications.

In a conference call it is sometimes difficult for the participating users to recognize immediately who is speaking. This is in particular a problem, in case there are many participating users in a conference call, while these participating users do not know each other very well.

The above cited RFC 1889 states that an example application is audio conferencing where a mixer indicates all the talkers whose speech was combined to produce the outgoing packet, allowing the receiver to indicate the current talker, even though all the audio packets contain the same SSRC identifier, i.e. that of the mixer.

In any sensible VoIP usage of a speech codec, however, the codec will send out Silence Descriptor (SID) frames enabling a comfort noise generation at the receiving end, as long as the respective conference participant is inactive, i.e. listening. Thus, all sources will always produce a signal that is transmitted to the conference call server 12. The conference call server 12 decodes VoIP flows received from each of the participants back to

speech or to SID frames for summation before encoding the outbound speech and SID frames that will be transmitted to the terminals 13. This implies that the SSRC identifiers of all terminals 13 are included by the mixer 15 into the CSRC list of the outgoing mixed RTP packets, and therefore it is impossible for the receiving terminals 13 to distinguish active from inactive participants. It has to be noted that it has also its benefits to include the SSRC identifiers of all participating terminals 13 in the CSRC list, e.g. in order to keep each participating user up to date about the number and identity of all other users participating in the conference.

SUMMARY OF THE INVENTION

It is an object of the invention to enhance the comfort of a user participating in a voice over IP conference call.

This object is reached according to the invention with a method for managing a packet switched, centralized conference call between a plurality of terminals, which comprising as a first step receiving at a conference call server data packets from all terminals participating in the conference call. These data packets include voice data or background noise information and an identifier associated to the respective terminal providing the voice data or the background noise information. In a second step, at least one terminal currently providing voice data, if any, is determined among the terminals participating in the conference call based on the received data packets. Obviously, in case none of the users participating in the conference call is talking for

a while, none of the terminals will provide voice data for a while, and no terminal can be determined which provides voice data. In a third step, the received voice data and the background noise information is mixed and inserted into new data packets together with at least one identifier associated to one of the terminals which were determined to provide currently voice data, if any. The identifier is included in a data packet in a way it can be distinguished from any other included information. This implies in particular that the at least one identifier can be distinguished from other possibly included identifiers which are not necessarily associated to terminals providing voice data. Finally, the new data packets are transmitted by the conference call server to terminals participating in the conference call.

The object of the invention is equally reached with a conference call server comprising means for realizing the proposed method.

In addition, the object of the invention is reached with a terminal which comprises means for participating in a centralized conference call, which means are suited to make use of the information transmitted according to the invention by a conference call server. The terminal comprising to this end means for receiving data packets transmitted by a conference call server. The data packets comprise mixed voice data and/or background noise information provided by terminals participating in the conference call and at least one identifier associated to a terminal that was determined in the conference call server to currently provide voice data, if any. Moreover, the terminal comprises means for recognizing in received data packets identifiers associated to terminals that

were determined in a conference call server to currently provide voice data. Further, the terminal comprises means for pointing out to a user an identification of terminals providing voice data based on recognized identifiers associated to terminals that were determined in a conference call server to currently provide voice data.

The invention proceeds from the idea that a conference call server can be designed to be able to distinguish between those participants of a conference call which are currently active, i.e. which provide voice data, and those which are currently inactive, i.e. which provide only background noise information. The invention further proceeds from the idea that a terminal can be designed to be able to point out to a user currently active participants of a conference call, in case it receives a corresponding information. Therefore, it is proposed that a conference call server performs a determination of the currently active participants of a conference call and that the server forwards a corresponding, distinguishable indication to the terminals participating in the conference call.

It is an advantage of the invention that it enables an improved user interface of a terminal, since transmitted information on the active conference participant can be presented to the user. The participants of the conference call can thus always identify the active speaker among all participants.

Preferred embodiments of the invention become apparent from the dependent claims.

The identifiers of active terminals can be transmitted by the conference call server in a variety of ways.

In a first alternative, the conference call server transmits in each combined data packet exclusively an identifier associated to those terminals, which are currently active. It is an advantage of this approach that the receiving terminals are able to indicate all active talkers to their users, even in case of multiple simultaneous talkers. With this approach, however, the receiving terminals are not able to keep their users up to date about all participants.

In a second alternative, the conference call server transmits in each combined data packet identifiers for all terminals participating in the conference, but in such a way that an identifier associated to an active terminal is always listed at a predetermined place in the list of identifiers, for example as the first element in the list. While this approach constantly provides up to date information about all conference participants, it does not allow to indicate more than one active terminal simultaneously. However, in a sensible discussion, especially over a telephone connection, only one participant will be talking at a time and this problem can be considered to be a minor one.

A third alternative is given by a refinement of the second approach. In this third approach, the conference call server always transmits again in each combined data packet identifiers for all terminals participating in the conference. The identifiers associated to the currently active terminals are listed at the beginning of the list of identifiers. In addition, some marker is inserted in

between the identifiers associated to currently active terminals and the identifiers associated to currently inactive terminals. This third approach combines the advantages of the first and the second approach, simply by introduction one additional value that has to be transmitted.

The identifier associated to a respective terminal might not be suited by itself to identify a transmitting terminal at a receiving terminal, like e.g. the randomly distributed SSRC identifier. In this case, preferably a mapping of the identifiers to a clear identification of the respective terminal is first transmitted from all possible transmitting terminals to the conference call server and further on to all possible receiving terminals. Then, each receiving terminal is able to map a later received identifier associated to a transmitting terminal to a corresponding identification of this terminal. The identification can be in particular a SIP address or a telephone number. The receiving terminal may also be able to further map the determined identification to another kind of identification. In case the identification is e.g. a SIP address or telephone number, the terminal may map this address or number to a name or an image stored in a directory of the receiving terminal.

In case all participants of the conference call are presented to the user of a terminal, the active participants can be pointed out to a user in any suitable manner.

The invention can be employed in particular, though not exclusively, in a system in which centralized conference calls are based on the RTP defined in the above cited RFC

1889. In this case, the data packets transmitted from the terminals to the conference call server and from the conference call server to the terminals are RTP packets. The identifiers of terminals transmitted by the conference call server in the combined RTP packets can be advantageously SSRC identifiers added to the CSRC list of the RTP header. In the third alternative presented for the transmission of identifiers by the conference call server, the employed marker can be for example the SSRC identifier associated to the conference call server. Since the SSRC identifier associated to the conference call server is transmitted anyhow in the SSRC field of the RTP header of each combined RTP packet, the receiving terminals have knowledge of this value and can use it for separating in the CSRC list active terminals from inactive terminals. In conventional applications, in contrast, the SSRC identifier associated to the conference call server is only included in the SSRC field of the outgoing combined RTP packets, not in the CSRC list, since the conference call server itself does not contribute to the combined RTP flow.

Each of the three alternatives presented for the transmission of identifiers by the conference call server complies with the current RTP specification and would not harm implementations that are not designed to make use of the special SSRC/CSRC handling.

A comprehensive embodiment of the method according to the invention implemented in an RTP based system advantageously comprises three parts. A first part consists in a mechanism for the terminals participating in a conference call to exchange RTP source identifiers and to map those identifiers to the respective identity

of each terminal or terminal user by means of RTCP SDES packets. A second part consists in a mechanism implemented in the conference call server for setting the CSRC field of RTP headers according to predefined rules. A third part consists in a mechanism implemented in the participating receiving terminals for mapping the identifiers in the CSRC field of the RTP packet headers to terminal or user identities, in order to enable a presentation of the identity of the currently active speaker to the users of the receiving terminals.

It is to be noted that the number of identifiers that can be transmitted by the conference call server to the participating terminals and/or the number of participants that can be presented by the receiving terminals may be limited to a predetermined value. According to the above cited RFC 1889, for example, the CSRC list is limited to a maximum number of 15 entries.

The invention can be employed in particular for Internet or UMTS packet switched voice conferencing. In case of UMTS, the information on the active participants can be shown e.g. on the screen of a mobile terminal.

BRIEF DESCRIPTION OF THE FIGURES

Other objects and features of the present invention will become apparent from the following detailed description considered in conjunction with the accompanying drawings, wherein:

Fig. 1 illustrates the principle of an RTP based, centralized conference call system;

Fig. 2 illustrates the structure of an RTP header; and

Fig. 3 shows a user interface of a terminal, which is making use of an embodiment of the method according to the invention.

DETAILED DESCRIPTION OF THE INVENTION

An embodiment of the method according to the invention will now be described with reference to figures 1 to 3.

The embodiment supports the management of VoIP conference calls and is implemented in an RTP based system which comprises a UMTS network 11, a conference call server 12 including an RTP mixer 15 connected to the network 11 and a plurality of terminals 13. The terminals 13 can be connected to the conference call server 12 via the UMTS network 11 by means of RTP tunnels 14. The system thus corresponds in general to the system illustrated in figure 1, which has already been described above.

For setting up a VoIP conference call in this system, the Session Initiation Protocol (SIP) is used as signaling protocol. SIP is used together with the Session Description Protocol (SDP) to send invitations to the called parties and to agree on the voice codecs etc. The users of the terminals 13 join the conference either by initiating the session themselves by sending the SIP INVITE message to the conference call server 12 or by replying to INVITE messages received via the conference call server 12.

At the beginning of an initiated conference session, the conferencing software in each terminal 13 sends RTCP SDES packets to the conference call server 12. These SDES

packets carry the SSRC identifier associated to the respective terminal 13 for this session and in addition in the SDES items field the SIP address or the phone number of the respective terminal 13. The conference call server 12 forwards the received SDES packets to each terminal 13 participating in the conference call. Based on the information in these SDES packets, the terminals 13 are prepared to map SSRC identifiers received during the conference session to corresponding SIP addresses or phone numbers.

When the conference session is active, all terminals 13 participating in the conference transmit RTP packets to the conference call server 12. The terminals 13 employ to this end a speech code, e.g. the AMR speech codec, in such a way that they transmit at a normal rate when there is speech at the input, i.e. when the user of the terminal 13 is talking, and with a reduced rate, when the source is silent, i.e. when the user of the terminal 13 is listening to the other participants. In the first case, the speech codec encodes voice data and transmits is in the payload of the RTP packet. In the latter case, the speech codec produces and transmits SID frames carrying a background noise estimate which is needed for the comfort noise generation at the receiver. In this case this receiver is the conference call server 12.

The RTP mixer 15 of the conference call server 12 decodes all incoming streams, in order to enable a summation of the decoded speech and an encoding of the combined speech. Based on the respectively employed data rate, the conference call server 12 obtains as a side information of the decoding process an indication on whether the decoded signal is speech or a background noise estimate.

Thereafter, the RTP mixer 15 of the conference call server 12 mixes the decoded voice data and the background noise estimates from all sources 13 together and assembles RTP packets with an encoded combined data flow. Each assembled RTP packet comprises an RTP header having a structure which corresponds to the structure illustrated in figure 2, which has already been described above. Thus, each RTP header comprises a field for an SSRC identifier and a field for a CSRC list.

The RTP mixer 15 inserts the SSRC identifier associated to the conference call server 12 for the current conference call to the SSRC identifier field of the RTP headers of the outbound RTP packets, since the conference call server 12 is the source for these RTP packets.

Moreover, the RTP mixer 15 includes the SSRC identifiers associated to those terminals 13 contributing to the combined RTP packets in the CSRC list of the RTP headers. Since all terminals 13 participating in the conference call always transmit RTP packets to the conference call server 12, either with voice data or with a background noise estimate, the CSRC list thus always comprises the SSRC identifiers for all participating terminals 13. The RTP mixer 15 takes care, however, that the SSRC identifiers which are associated to the actively participating terminals 13 are included as first elements in the CSRC list.

Additionally, the RTP mixer 15 inserts also the SSRC identifier associated to the conference call server 12 to the CSRC list. More specifically, the SSRC identifier associated to the conference call server 12 is included

as a marker between the SSRC identifiers associated to the active terminals 13 located at the beginning of the CSRC list and the SSRC identifiers associated to the inactive terminals 13 located at the end of the CSRC list.

The conference call server 12 then forwards the composite flow to each participating terminal 13.

The terminals 13 receive the RTP packets transmitted by the conference call server 12 via the UMTS network 14 and retrieve the SSRC identifiers included in the respective CSRC list of the headers of the RTP packets. Based on the mapping information received earlier, the terminals 13 then determine the SIP addresses or the phone numbers corresponding to the SSRC identifiers retrieved from the CSRC list. The terminals 13 do not perform such a mapping for the SSRC identifier which is associated to the conference call server 12. This SSRC identifier is recognized by the terminals 13 based on the identical SSRC identifier included in the SSRC identifier field of the RTP header. The terminals 13 further determine names which are associated in their internal address directories to the determined SIP addresses or phone numbers, as far as available. The determined names are then presented to a respective user on the display of the terminals 13 in form of a list.

Figure 3 shows an embodiment of such a display 31, which presents beside other information and options a list 32 with the names of users participating in an on-going conference call.

In addition, the terminals 13 determine all those SSRC identifiers in the CSRC list which are listed before the SSRC identifier associated to the conference call server 12. The names which were determined for those SSRC identifiers belong to currently active participants and are pointed out in the presented list 32 on the display 31. In the example of figure 3, a special speaker indicator icon 33 is employed for indicating the participants who are currently speaking. In the presented situation, only one participant is currently talking, and a speaker indicator icon 33 is located next to the corresponding name, "Saimi", in the list 32.

Thus, the user of a terminal 13 is always able to see an identification of all users participating in the conference call, and to distinguish the currently speaking participants from the inactive participants.

It is understood that the described embodiment constitutes only one of a variety of possible embodiments of the invention.

Claims

1. Method for managing a packet switched, centralized conference call between a plurality of terminals (13), said method comprising at a conference call server (12):
   - receiving data packets from all terminals (13) participating in said conference call, which data packets include either voice data or background noise information as well as an identifier associated to the respective terminal (13) providing said voice data or said background noise information;
   - determining based on said received data packets at least one terminal (13) currently providing voice data, if any, among said terminals (13) participating in said conference call;
   - mixing said received voice data and said received background noise information and inserting said mixed data into new data packets together with at least one identifier associated to one of said terminals (13) which were determined to provide currently voice data, if any, such that said at least one identifier can be distinguished from any other information included in said data packets; and
   - transmitting said new data packets to terminals (13) participating in said conference call.

2. Method according to claim 1, wherein said identifiers associated to said terminals (13) are identifiers

associated randomly to said terminals (13) for said conference call, said method comprising as preceding steps receiving at said conference call server (12) control packets from said terminals (13) participating in said conference call, said control packets including a mapping of an identifier associated a respective terminal (13) to an identification of said terminal (13), and forwarding said mapping in control packets from said conference call server (12) to said terminals (13) participating in said conference call.

3. Method according to one of the preceding claims, wherein said conference call server (12) transmits in said new data packets exclusively identifiers associated to terminals (13) which were determined to provide voice data.

4. Method according to claim 1 or 2, wherein said conference call server (12) includes in said new data packets identifiers associated to terminals (13) currently providing voice data as well as identifiers associated to terminals currently providing background noise information, at least one identifier associated to a terminal (13) which was determined to provide voice data being included in said data packet at a predetermined position among all included identifiers.

5. Method according to claim 1 or 2, wherein said conference call server (12) includes in said new data packets identifiers associated to terminals (13) currently providing voice data as well as identifiers associated to terminals (13) currently providing

background noise information, wherein at least one identifier associated to one of said terminals (13) which were determined to provide voice data, if any, is included in said data packets at a predetermined position among all included identifiers, and wherein identifiers associated to terminals (13) which were determined to provide voice data are separated by a marker from included identifiers associated to other terminals (13).

6. Method according to claim 5, wherein said marker corresponds to an identifier associated to said conference call server (12).

7. Method according to one of the preceding claims, wherein said conference call is based on the Real-time Transport Protocol (RTP), wherein said data packets are RTP packets, wherein said identifiers associated to said terminals (13) are Synchronization Source (SSRC) identifiers, and wherein said identifiers are included by said conference call server (12) in said new data packets to a field provided in a packet header for a Contributing Source (CSRC) list.

8. Method according to one of the preceding claims, further comprising receiving said new data packets transmitted by said conference call server (12) at a terminal (13) participating in said conference call and pointing out an identification (32,33) of at least one terminal (13) determined to provide voice data to a user based on an identifier included in said received new data packets.

9. Conference call server (12) comprising means for managing a centralized conference call between a plurality of terminals (13), said means including means (15) for realizing the steps of the method according to one of claims 1 to 7.

10. Terminal (13) comprising means for participating in a centralized conference call, said means including
   - means for receiving data packets transmitted by a conference call server (12), which data packets comprise mixed voice data and/or background noise information provided by terminals (13) participating in said conference call and at least one identifier associated to a terminal (13) that was determined in said conference call server (12) to currently provide voice data, if any;
   - means for recognizing in received data packets identifiers associated to terminals (13) that were determined in a conference call server (12) to currently provide voice data; and
   - means for pointing out to a user an identification of terminals (13) providing voice data based on recognized identifiers associated to terminals (13) that were determined in a conference call server (12) to currently provide voice data.

Abstract

The invention relates to a method for managing a packet switched, centralized conference call between a plurality of terminals 13. In order to enable an enhancement of the user comfort, it is proposed that the method comprises at a conference call server 12 receiving data packets from all terminals 13. Based on these data packets, then at least one terminal 13 currently providing voice data is determined. In a next step, the data received in the data packets is mixed, and the mixed data is inserted into new data packets together with at least one identifier associated to one of the terminals 13 which were determined to provide voice data, such that the at least one identifier can be distinguished from any other information in the data packets. Finally, the new data packets are transmitted to terminals 13 participating in the conference call. The invention relates equally to a corresponding server and to a corresponding terminal.

For publication: Figure 3

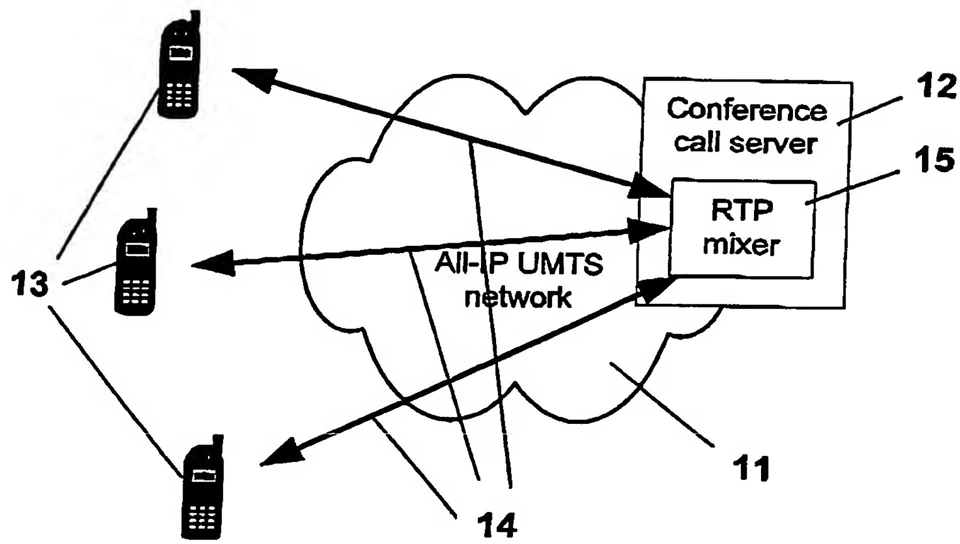
Conference call server
RTP mixer
All-IP UMTS network
12
15
13
11
14

FIG. 1

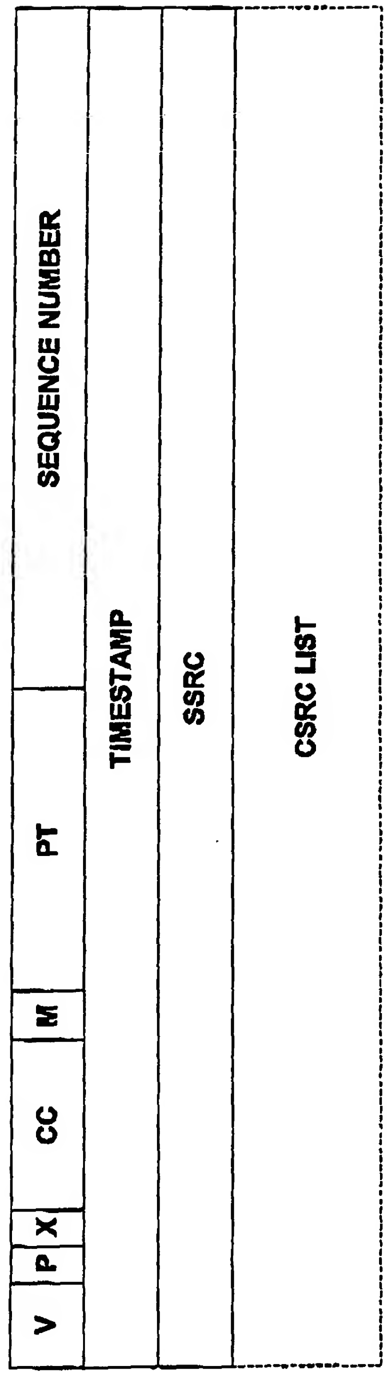

FIG. 2

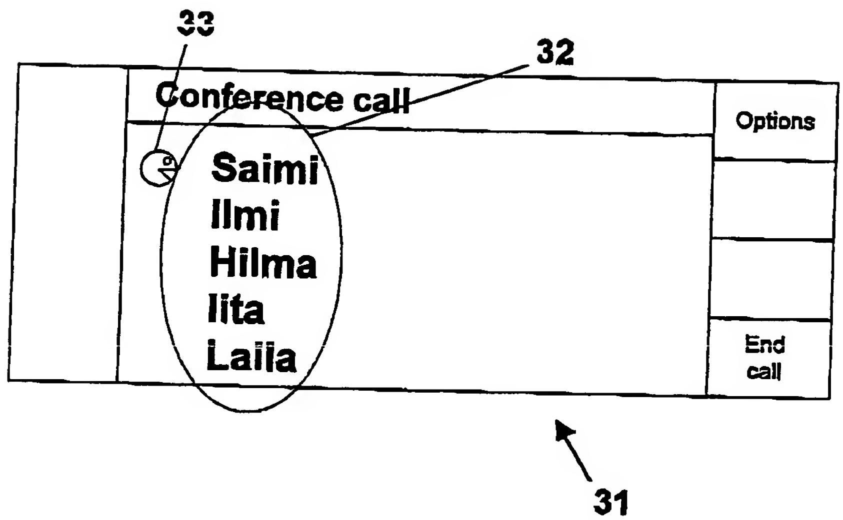
33
32
Conference call
Options
Saimi
Ilmi
Hilma
Iita
Laila
End call
31

FIG. 3